The Session Initiation Protocol

N. C. State University

CSC557 ♦ Multimedia Computing and Networking

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Lecture # 25

"Roadmap" for Multimedia Networking

1. Introduction
   - why QoS?
   - what are the problems?
2. Basic operations
   - jitter buffers (at hosts)
   - task scheduling (at hosts)
   - packet shaping (at hosts)
   - packet dropping (at routers)
   - packet scheduling (at routers)
3. Types of service
   - Integrated Services (IntServ) and Resource Reservation Protocol (RSVP)
   - Differentiated Services (DiffServ)
4. Application-level feedback and control
   - Real-time Protocol (RTP), Real-time Control Protocol (RTCP)
5. Application signaling and device control
   - Session Description Protocol (SDP)
   - Real-time Streaming Protocol (RTSP)
   - **Session Initiation Protocol (SIP)**
   - Media Gateway Control Protocol (MGCP)
6. Routing
   - Multi-protocol Label Switching (MPLS)
   - multicasting

Today's Lecture

Voice Over IP Goals

1. Recreate all the functionality the current public switched telephone network (PSTN) offers

2. Plus, offer new services and features not currently available

VoIP Components

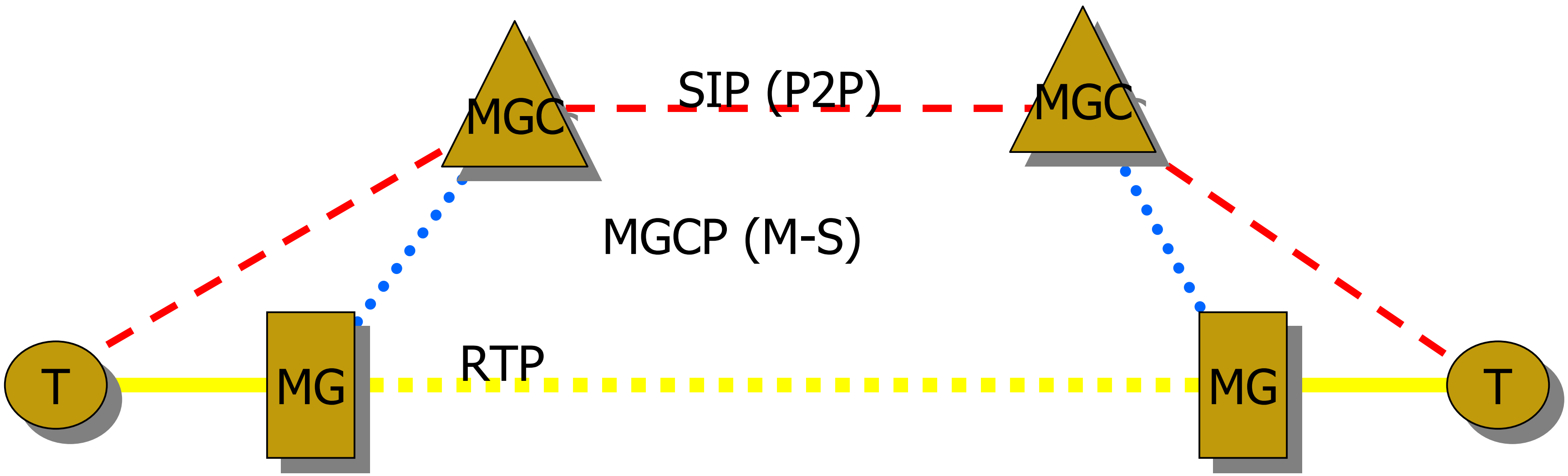
- Terminals (endpoints)
  - Computers, conventional telephones, "softphones"
- Media gateways (MG)
  - i.e., "translators"
  - for conversions between compression formats, networking technologies, media transport standard
  - a gateway may be part of a single telephone
- Media gateway controller
  - Also called call manager, call controller, or call server
  - 2 main functions: call routing, and gateway/terminal control
- IP routers
  - for transporting the datastream

Entities (cont.)

- Signaling gateway (SG)
  - For conversions between signaling standards
  - Often accomplished via encapsulation (wrap one protocol inside another)
- The signaling path vs. the media path

Master-Slave or Peer-to-Peer?

- Master-slave (M-S) relationship is easiest to manage, easiest to recover from faults
- Peer-to-peer (P2P) is most flexible, most distributed (load sharing)



An all-IP phone network (i.e., no signaling conversions)

Session Initiation Protocol (SIP)

- Used to initiate, control, and terminate telephone calls and other services (voice and otherwise) over the Internet
  - end-to-end call signaling, possibly involving media gateway controllers (MGC) between the end points
  - "SIP is a much simpler protocol than H.323, but is at least as functional‟
- Properties
  - peer-to-peer
  - fully distributed
  - lightweight
  - text-based (human readable)
- Protocol implementation is modelled on HTTP

SIP Key Features

- Functionality
  - SIP is complete for setting up point-to-point or multiparty multimedia calls
  - Extensive call handling capabilities
  - Media capabilities are negotiated
- Flexibility
  - URLs are used for addresses (i.e., to locate the callee)
  - Users can move to new locations and access their full telephony features from anywhere
  - Users can define what response they want to give when contacted (availability, etc.)

SIP Proxies (i.e., Call Servers)

- Users may require the use of SIP proxies, or call servers, to set up and maintain the call
  - find the other party
  - configure the media gateways
  - do user authentication, authorization, and billing
- SIP servers are stateless
  - means any information about the current state of the call is stored in the endpoints (gateways, terminals), rather than in the server
  - reason: to simplify the design of the servers, improve their scalability
- UDP is normally used for SIP transport
  - messages may be lost, have to be retransmitted by application

Call "Routing"

- Translation steps
  1. user keys in phone number
  2. SIP proxy translates phone number into URL
  3. URL contains name of host to contact
  4. DNS resolves host name to IP address
  - now the call can be routed…
- Allows great flexibility in…
  - directing phone calls, based on who is initiating the call
  - moving phones and services around, without impacting the users of the service

Call Routing Example (with *Relay*)

1. Jon initiates a call to eve@isi.edu by issuing INVITE message
   - (could also have been a phone number)

2. DNS looks up a SRV record for the SIP server (proxy) at isi.edu; proxy is sip.isi.edu

3. The caller invitation is sent to sip.isi.edu

4. sip.isi.edu has a database indicating Eve is served by another SIP server sipgw.cs.isi.edu

5. sip.isi.edu relays the request to sipgw.cs.isi.edu

6. sipgw.cs.isi.edu has database indicating how to reach Eve
   — this database is updated whenever Eve registers a new location

Example (cont.)

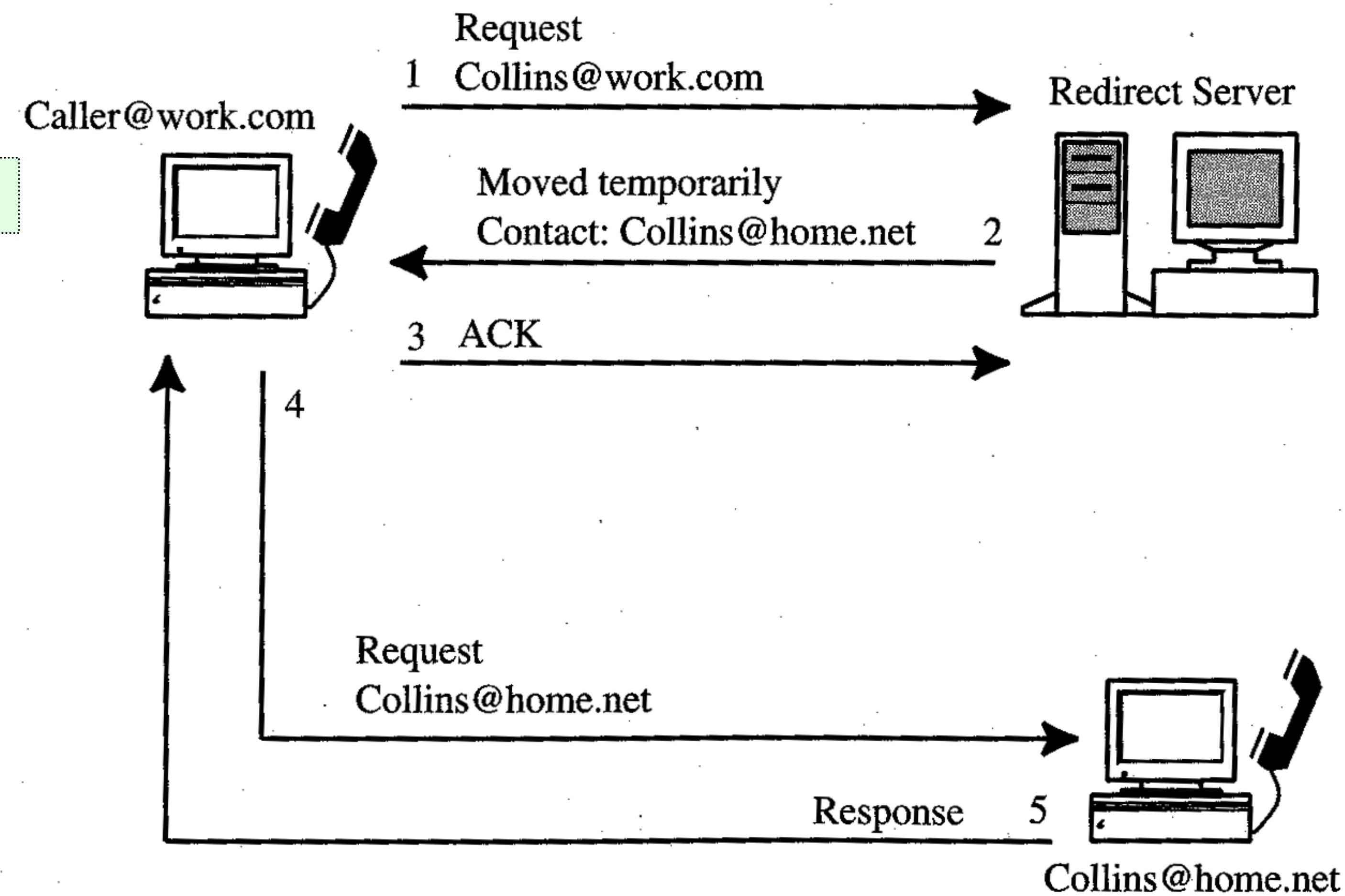
7. sipgw.cs.isi.edu relays the invitation to Eve at her current location, perhaps through a media gateway

- Note A: SIP servers (proxies) are optional, not required
  - users can place phone calls directly to each other if they already know what hostname / IP address to use
- Note B: rather than relaying requests to the callee, proxies may reply directly to the caller
  - the caller should then contact the callee directly
  - this is called a *redirect*

Ex.: Call Routing (with *Redirect*)

Figure 5-4
SIP redirect operation

Source: Carrier Grade Voice Over IP, D. Collins, McGraw-Hill, 2001



Path "Marking"

- An invitation is sent to a callee through one or more proxies
- As each proxy encountered on the Invite path, it adds a “via" line to the header
  - contains the identity of this proxy
- The callee response can then be returned to the caller along this same path, in reverse order
  - allows proxies to be notified of all messages between caller and callee
  - Each proxy removes its “via” information from the reponse message
- No “state” is installed at any server; all information about the path is contained in the messages

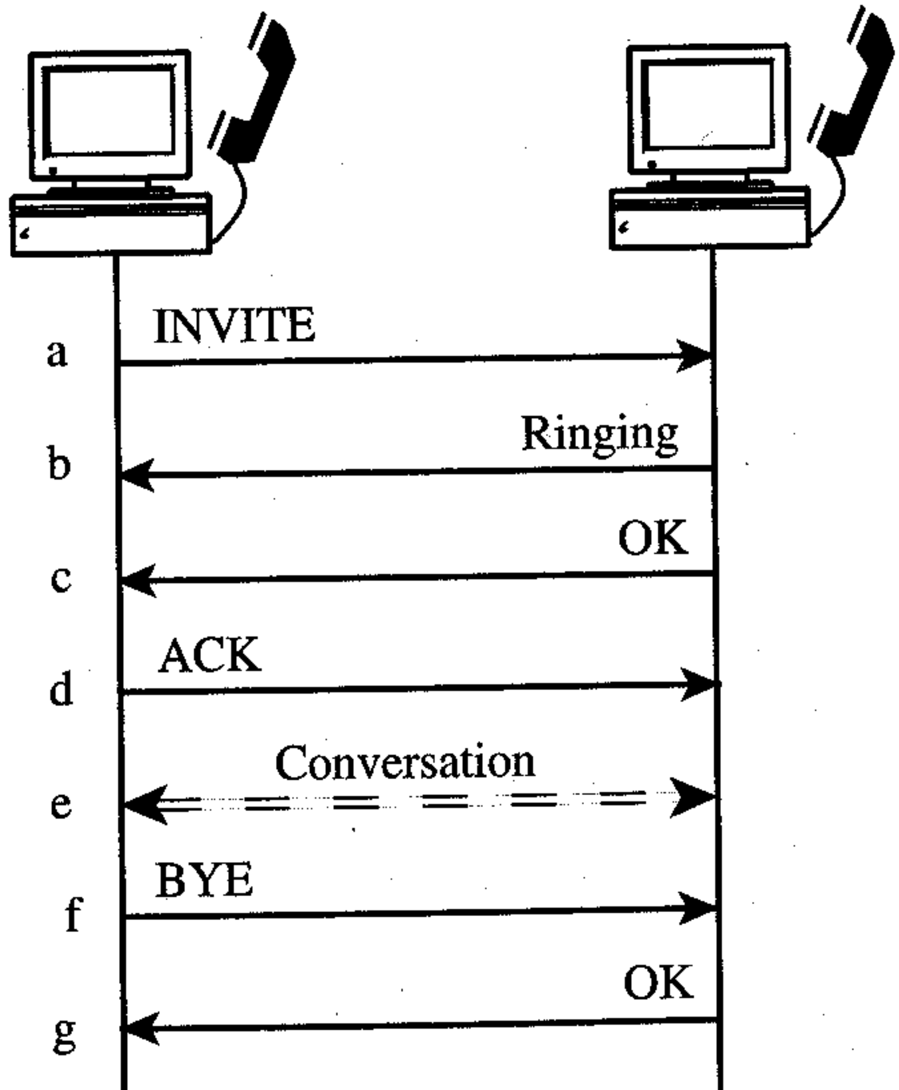
Mandatory Fields in Every Message Header

- Call ID
- Call Sequence #
  - CSEQ = a transaction in the call signaling
- From
- To
- Via (if proxied)

Example: "Normal" SIP Call

Figure 5-5
Example of an SIP call

Source: Carrier Grade Voice Over IP, D. Collins, McGraw-Hill, 2001



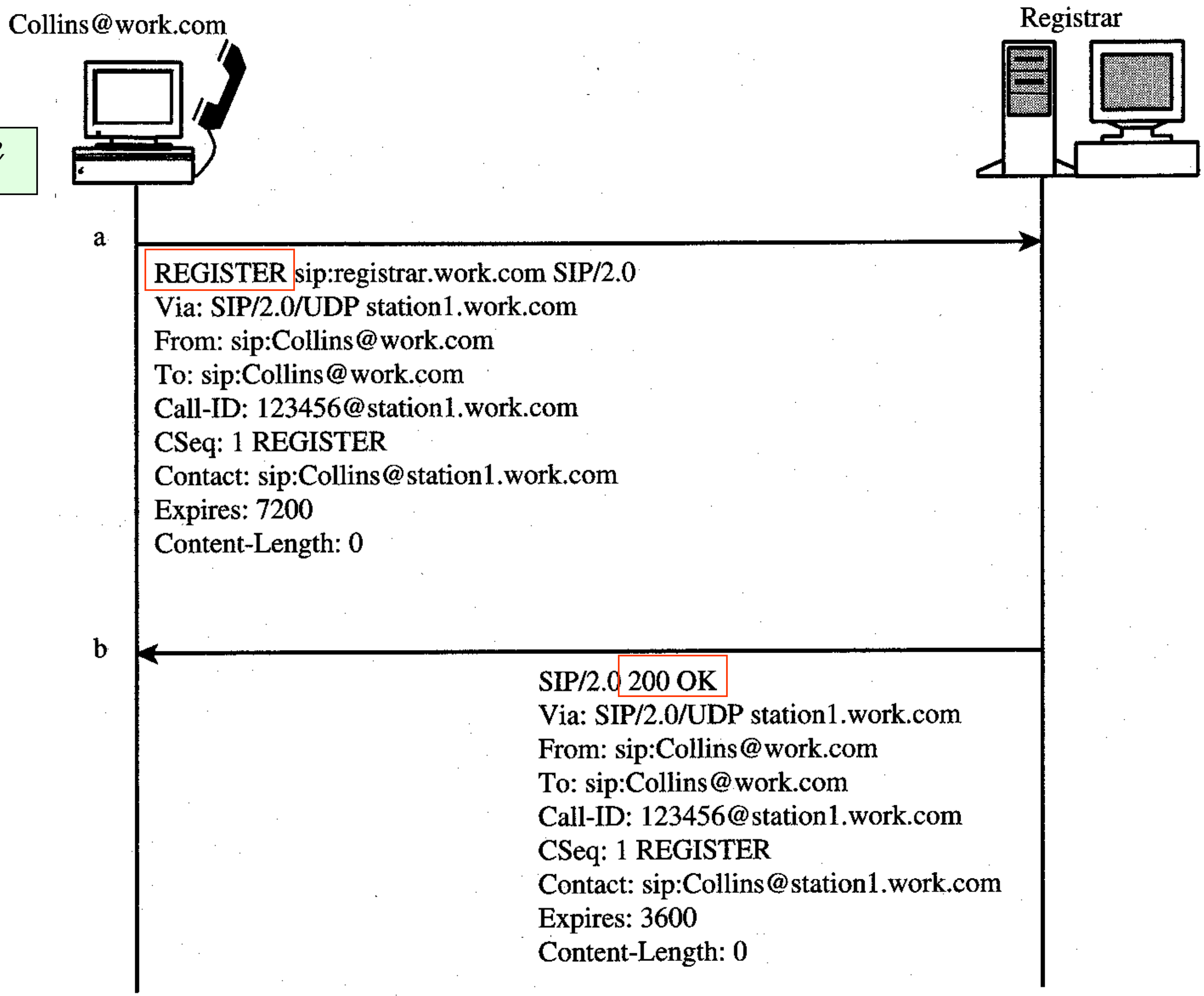
REGISTER Request Method

- Purpose: register the current location of a user
- Who register with?
  - multicast to the well-known "all SIP servers" multicast address "sip.mcast.net" (224.0.1.75)
- What happens if user is registered as being at multiple locations? Several choices:
  1. contact them all at once, wait for the first to respond, ignore other responses
  2. contact them one at a time until get successful response
  3. redirect server sends them all the locations; up to caller to decide what order to try them

Registration: Detailed Example

Figure 5-8
SIP registration

Source: Carrier Grade Voice Over IP, D. Collins, McGraw-Hill, 2001



INVITE Request Method

- Purpose: invite the callee to join in a session (conversation)
  - issued to initiate a call
  - or, to change the state of a call
    - e.g., put someone on hold while accepting another call
- Includes
  - From (caller identification)
  - To (callee identification)
  - Call Sequence #
  - SDP description of call parameters
  - Via (if proxied)
- All *except* SDP description are mandatory

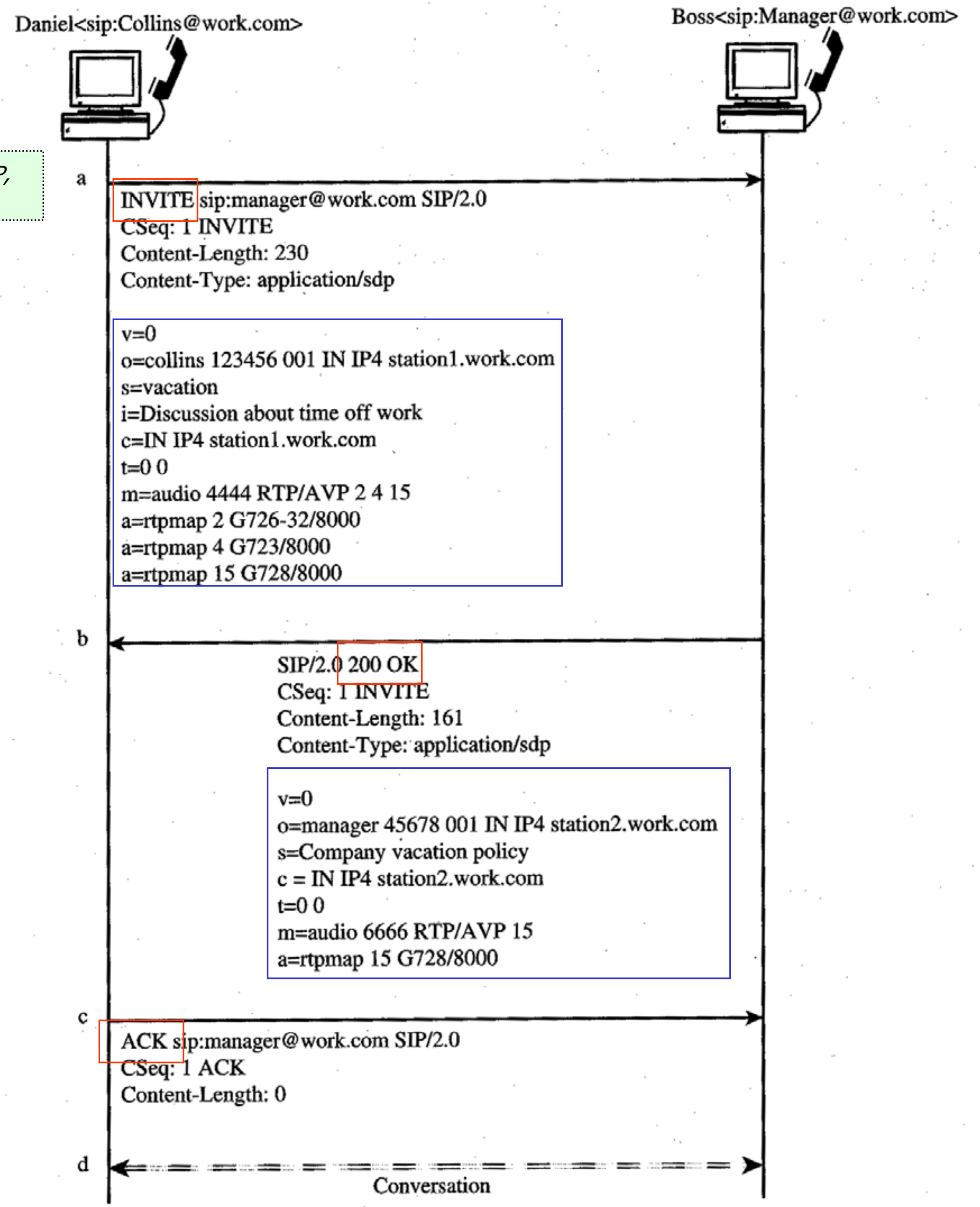
Media Negotiation

- The invitation (from caller to callee) suggests the type of media sessions to establish
  - using SDP description
- The response (from callee to caller) indicates type of media sessions that are possible / acceptable
- Caller "re-invites" with the commonly-agreed set of media functions

Including Session Description in Invitation

Figure 5-15
SDP inclusion in SIP messages

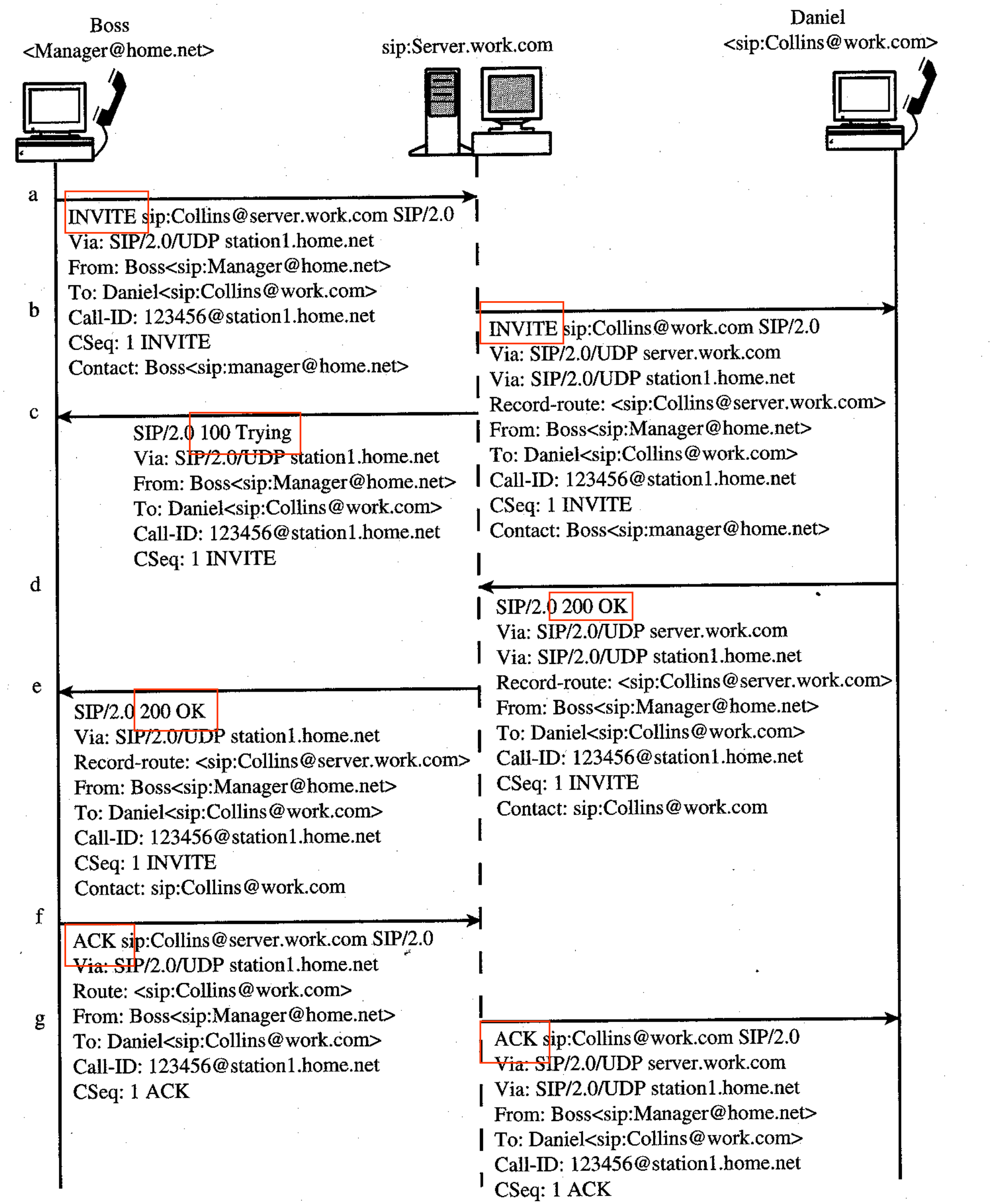
Source: Carrier Grade Voice Over IP, D. Collins, McGraw-Hill, 2001



Ex: Use of a Proxy Server

Figure 5-12
Application of a proxy server

Source: Carrier Grade Voice Over IP, D. Collins, McGraw-Hill, 2001



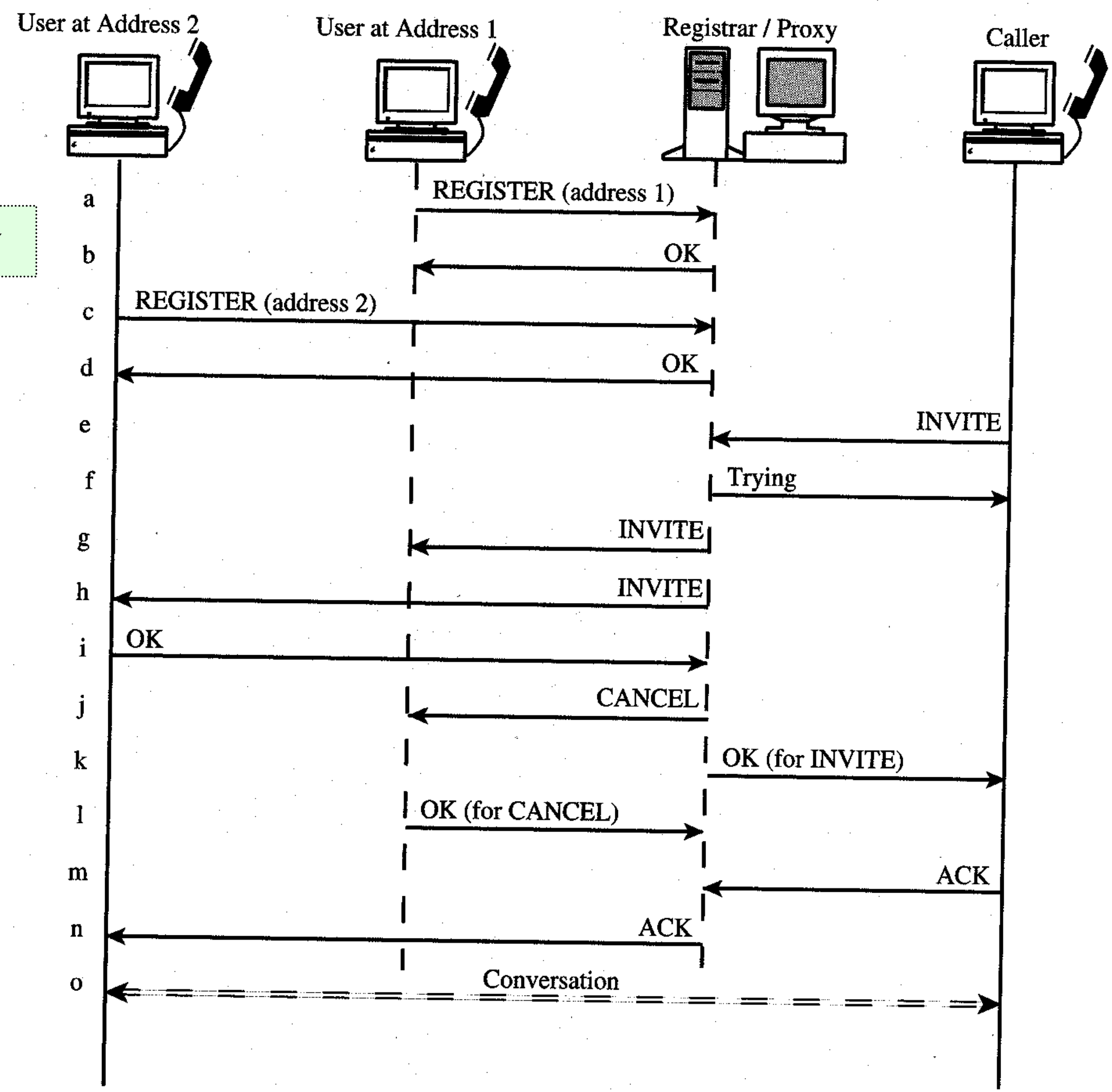
CANCEL Request Method

- Purpose: terminate incomplete call requests
  - no effect on established calls

Ex.: Try All Locations at Once

Figure 5-7
Multiple registrations enabling a "one-number" service

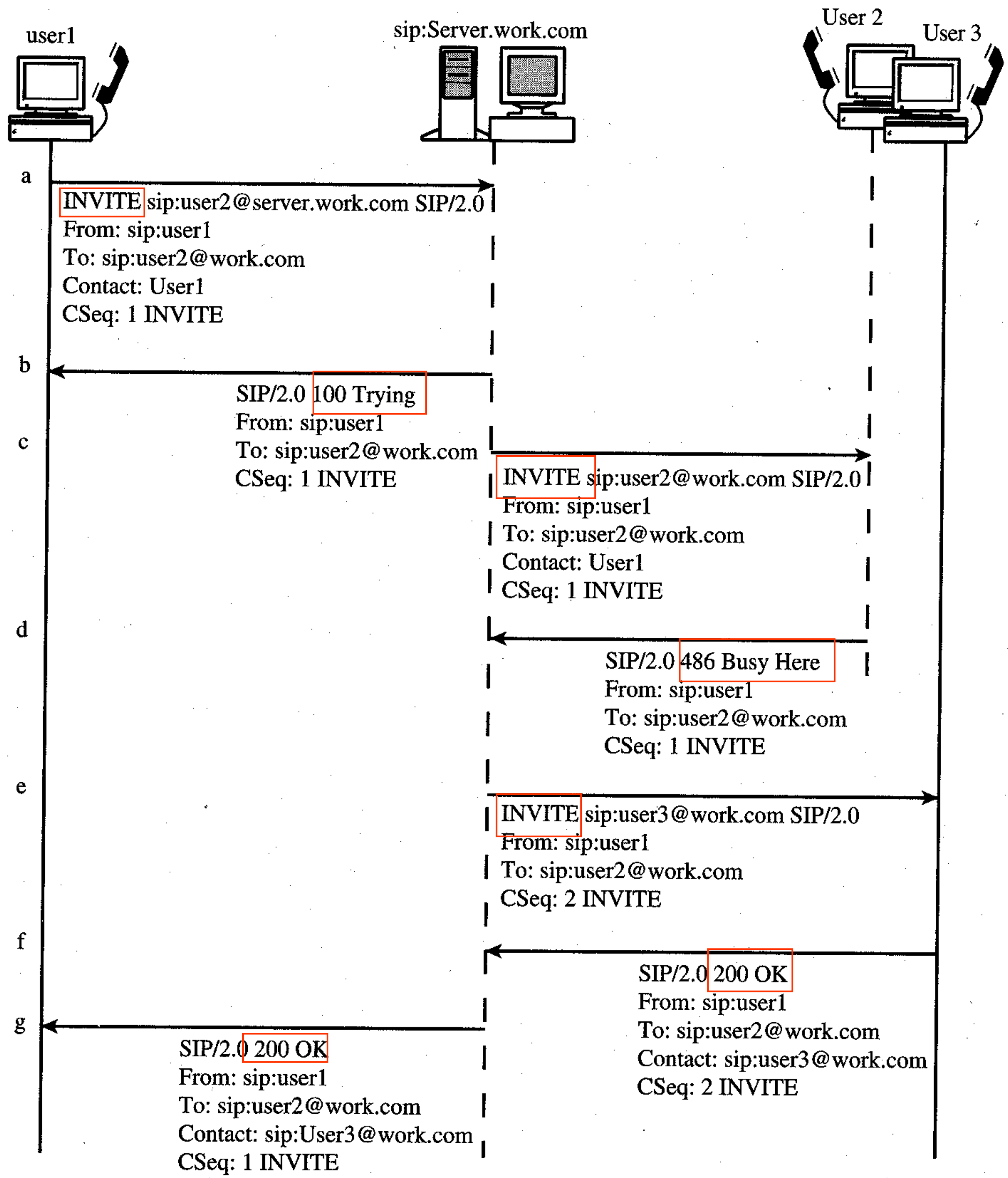
Source: Carrier Grade Voice Over IP, D. Collins, McGraw-Hill, 2001



Ex.: Call Forwarding

Figure 5-18
Call forwarding on busy

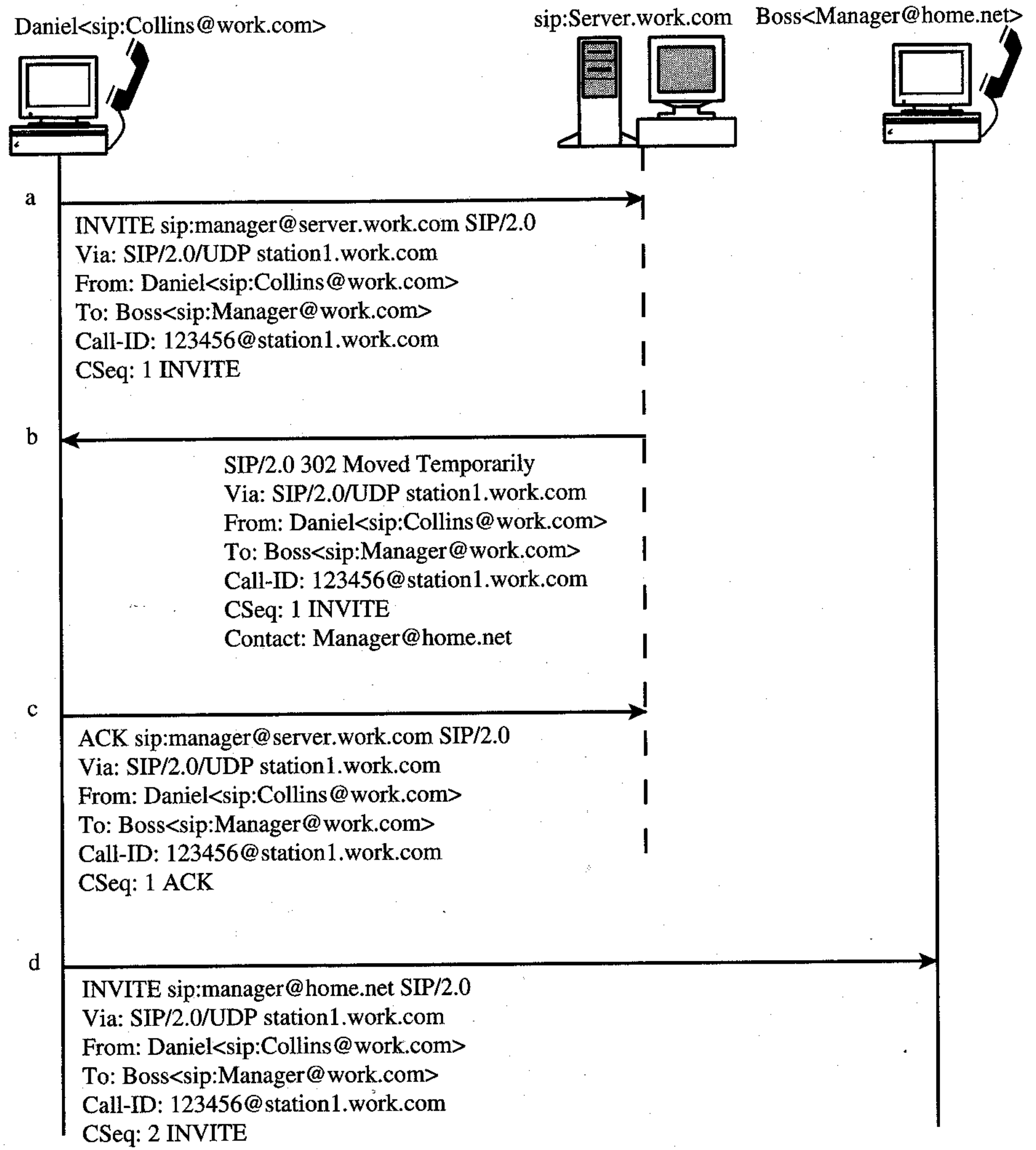
Source: Carrier Grade Voice Over IP, D. Collins, McGraw-Hill, 2001



Ex: Redirect

Figure 5-11
Application of a redirect server

Source: Carrier Grade Voice Over IP, D. Collins, McGraw-Hill, 2001



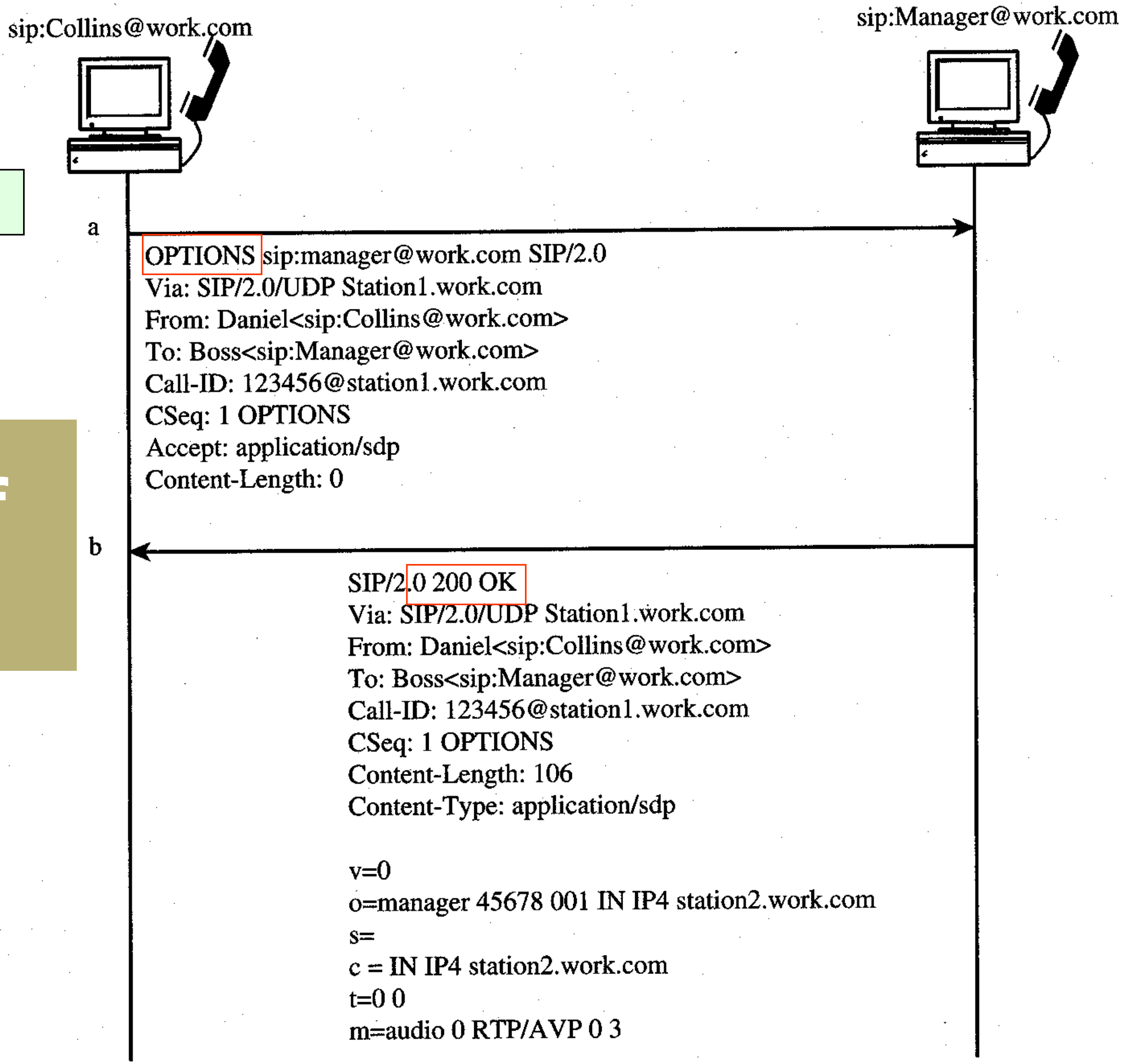
OPTIONS Request Method

- Purpose: request / discover the capabilities of the receiver
- Includes
  - which media types are supported
    - format: SDP
  - which request methods are supported

Ex.: Use of Options

Figure 5-16
Usage of the OPTIONS method

Source: Carrier Grade Voice Over IP, D. Collins, McGraw-Hill, 2001



BYE Request Method

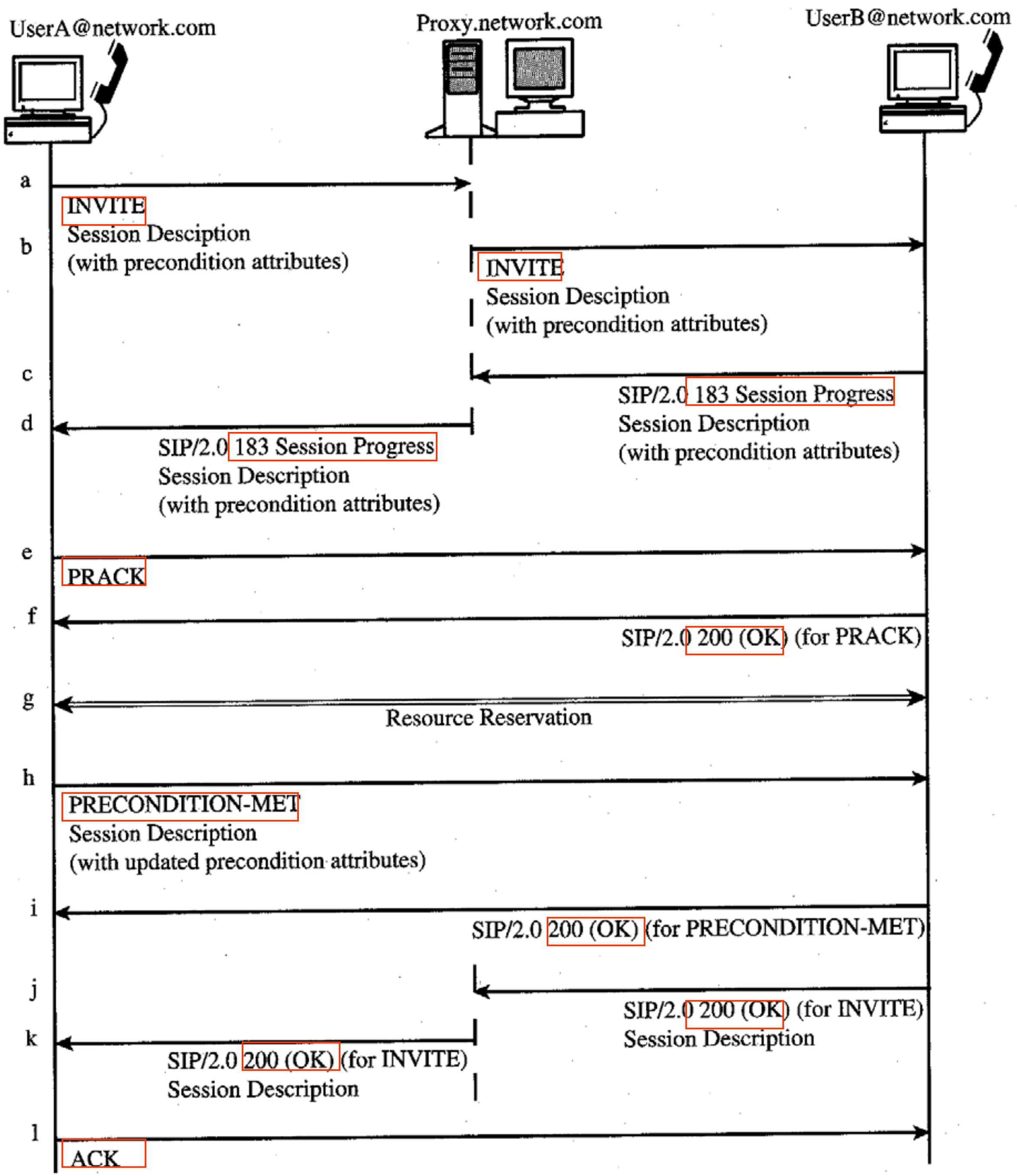
- Purpose: terminate call or call request
  - also, terminate the media flow
- Not required to wait for a response

Integration of Resource Reservation and Signaling

- "Don't ring the phone until resources have been reserved"
  - Precondition-met means resources have been reserved

Figure 5-22
Integration of resource reservation and SIP signaling

Source: Carrier Grade Voice Over IP, D. Collins, McGraw-Hill, 2001



SIP Response Codes

- 100…199: Call status
- 200…299: Success
- 300…399: Redirection
- 400…499: Client error
- 500…599: Server error
- 600…699: Global failure

1XX -- Information about Call Status

- 100 Trying
- 180 Ringing
- 181 Call forwarding
- 182 Queued for service

2XX -- Success Indicator

- 200 OK

3XX -- Redirection to Another Proxy

- 300 Address ambiguous (and possible matches returned)
- 301 Moved permanently (and here is the new address or addresses to try)
- 302 Moved temporarily (here is the temporary address for call forwarding)
- 305 Callee cannot be contacted directly (must be contacted through proxy)

4XX -- Client Did Something Incorrectly

- 400 Request syntax error
- 401 Unauthorized / unauthenticated to make call
- 402 Called user owes money!
- 403 Forbidden request, do not reattempt
- 404 Called user not found
- 405 Called user does not support options indicated
- 408 Server cannot respond within time indicated in request header
- 409 Conflict between requests (such as duplicate registrations)

4XX (cont'd)

- 410 No forwarding address available
- 420 Server does not understand SIP extension being used
- 480 Called party temporarily unavailable
- 481 Cancel or bye for non-existent call
- 482 Loop in routing detected
- 483 # of hops in route exceeds limit
- 485 Ambiguous address (with possible addresses offered)
- 486 Callee busy or unwilling to accept call (may indicate time to try again)

5XX -- Server Error

- 500 Internal server error
- 501 Service not implemented
- 503 Service temporarily unavailable
- 504 Server timed out
- 505 SIP version not supported

6XX -- Global Failure (modify request before resending)

- 600 Called party is busy
- 603 Called party declined call
- 604 Called user does not exist
- 606 Media choice unacceptable / unsupported

Headers

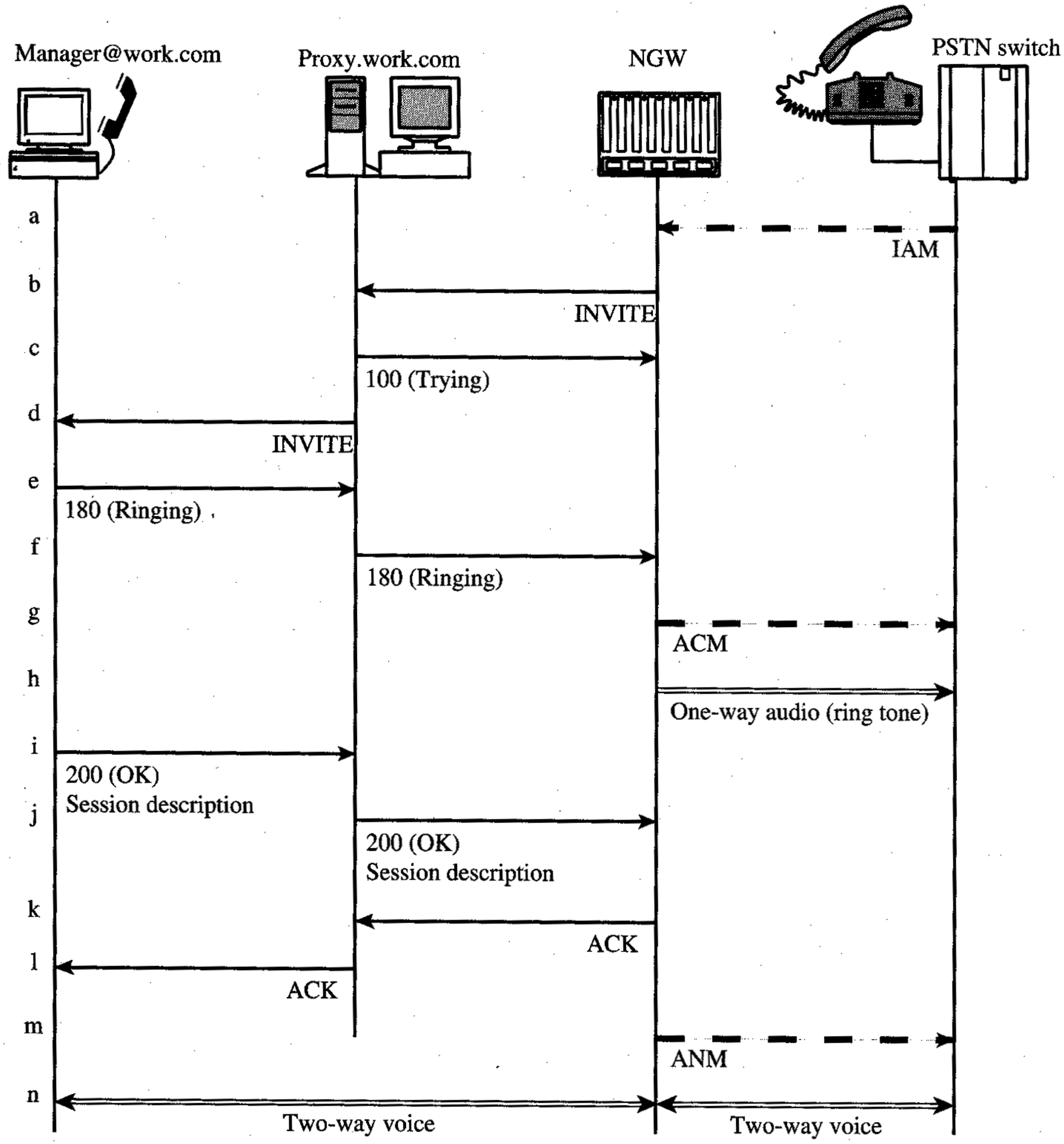
- Allowed categories of information that can be contained in a SIP message
- Approximately 40 defined; see RFCs for specification

Figure 5-24
PSTN to SIP call

Source: Carrier Grade Voice Over IP, D. Collins, McGraw-Hill, 2001

PSTN to SIP Call

- Signaling used for "conventional" (PSTN) phone calls: SS7/ISUP
- Establishing calls between phone network and SIP phones requires translation between SIP and ISUP

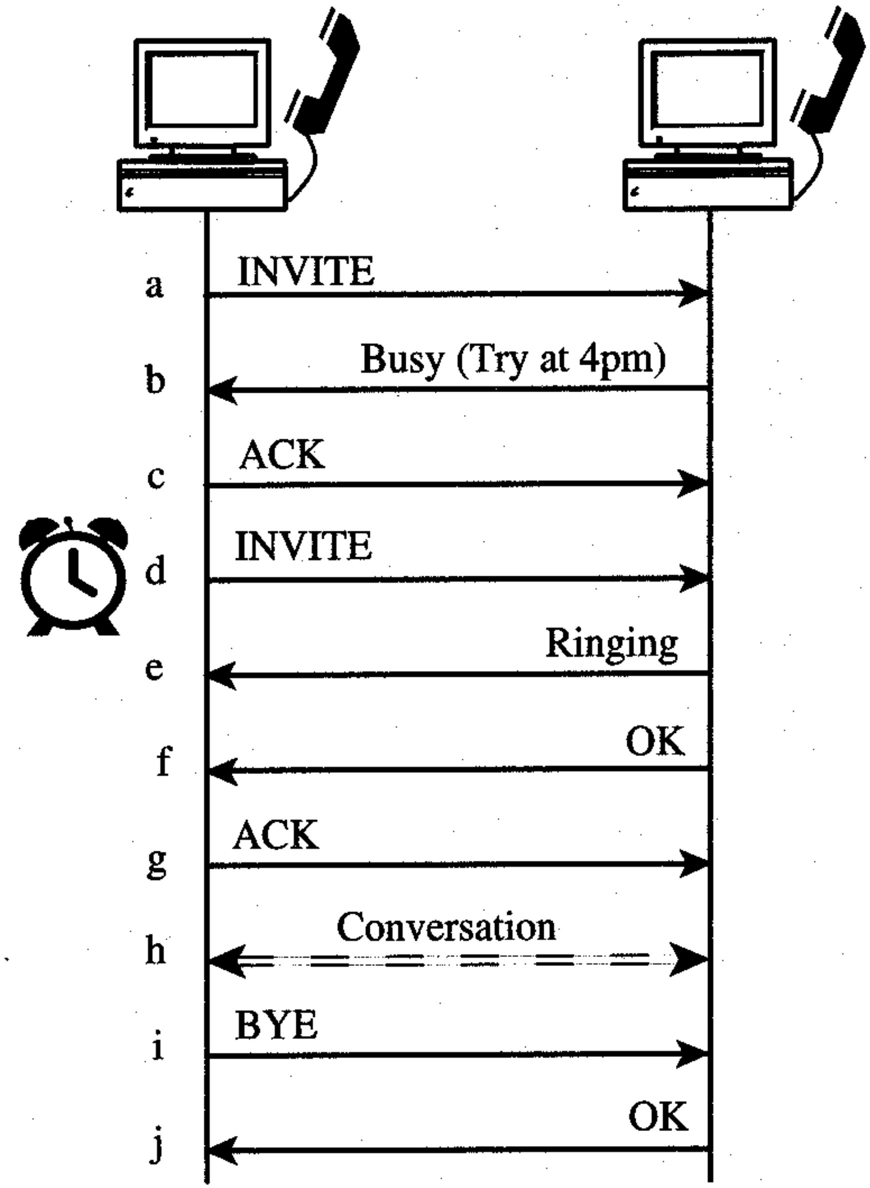


SIP = SERVICE Initiation Protocol

Figure 5-6
Example of an SIP-enabled service

Source: *Carrier Grade Voice Over IP, D. Collins, McGraw-Hill, 2001*

- Example: automatic ringback at a time specified by callee



Sources of Info

- Recommended Books
  - Online!: J. Crowcroft et al., *Internetworking Multimedia*, 1999
  - D. Collins, *Carrier-Grade Voice over IP*, 2001
    - Chapter 5
- Other books
  - O. Hersent et al., *IP Telephony*, 2000
    - Chapter 2
  - B. Douskalis, *IP Telephony*, 2000
- RFCs
  - SIP: Session Initiation Protocol, RFC 2543
  - http://www.ietf.org/internet-drafts/draft-ietf-sip-rfc2543bis-01.txt